detour. As a result, if branch 195c is 125 feet, the energy of the reflected signal in Fig. 2 will be at least 21 dB (17.5 dB due to the detour plus 3.5 dB contributed by split 199) below the energy video signal that follows the direct path of branches 195a, 195b. The reflection loss at termination 197 may make the reflected signal level slightly lower. Because interference suppressed as much as 40dB can still affect an AM (amplitude modulated) video signal, the reflected signal would cause multipath interference if the transmitted signal is AM encoded.

Splits and connected telephone devices encountered along a detour, however, can prevent multipath interference from occurring. Because it is unusual to find a "clean" 125 foot branch with no splits or telephone devices, this is an important property. The mechanism by which splits and connected telephone devices can prevent multipath interference is explained in the following paragraphs.

The routing and attenuation of reflected signal energy is very different if the dashed line labeled sub-branch 198 represents a secondary branch connected to branch 195c, whose length is comparable to or larger than a quarter of a wavelength. In this event, the split 198′ created by sub-branch 198 causes the reflected signal analyzed above to lose 3.5 dB while passing from split 199 to termination 197, and another 3.5 dB while returning from termination 197 after the reflection. Although the lost energy stays on the network in the form of reflected signals that will ultimately find their way back to receiver 196, these "secondary" reflections will have different delays or offsets. This means that their energy will not add coherently and the combined effect of the various reflected signals will be dominated by the effect of the strongest reflected signal. Termination 197, moreover, may include a connected telephone device which can further attenuate the reflected signal.

Because 125 foot branches with no significant

secondary branches and no connected telephones are unusual, multipath interference is not likely at 30 MHz or above. At 10 MHz, however, the attenuation of telephone wiring is approximately 2.5 dB per 100 ft. The loss due to attenuation in this case is only 6dB. This means the energy differential between reflected and direct signals is less at 10 MHz than at 30 MHz, making multipath effects more likely at the lower frequency.

To avoid multipath interference, the following solution, embodied in splitter 161 (Figs. 1 and 1A), is disclosed. Reflections at the termination (i.e., a telephone jack) of a long branch are suppressed by altering the impedance of the termination to match that of the wiring at the frequencies of transmission. Video signals incident at such a termination will not reflect but will behave as if the conductive path continues without end. Thus, video signals and other energy presenting at this termination will be removed from the wiring network.

In some circumstances, the removal of energy by these terminations can have a detrimental effect. Consider, for example, the case where a main transmission path has 10 short stubs connected to it, each of which provides a port for connection of telephones. Terminating each of these in this manner would remove 3.5 dB of energy at each stub, a total reduction of 35dB. Because the ports are connected via short stubs, furthermore, they are not likely to cause multipath problems. Thus, termination of the stubs would be unwise in this case. (Use of low-pass filters to prevent draining of high frequency energy by connected telephones, however, is still very useful.)

In general, branches should be terminated only when multipath interference would otherwise result from a reflected signal. In the case of AM video, interference only occurs when a signal traverses a reflected path longer than approximately 200 feet, and is received at a level within 40dB of the level of the strongest signal.

Figs. 1 and 1A show a method of altering the termination of a telephone jack at frequencies above voiceband. Video signals (i.e., RF signals as produced by, e.g., any of the transmitters disclosed herein) incident at network port 168 on the red-green wire pair are applied equally to (i.e., split between) low-pass filter 162a and high-pass filter 164. The RF signals pass through filter 164 to switch 165, shown in its normally closed position in Fig. 1. Switch 165 is actuated by arm 165a, the position of which is a function of whether a RF receiver (such as any of the receivers described in U.S. Patent No. 5,010,399 application or elsewhere in this application) is connected to RF port 167 or, alternatively, whether RF port 167 is "open."

Arm 165a is pivotally mounted at RF port 167 and biased by spring 165b to maintain switch 165a in the normally closed position whenever RF port 168 is "open" (i.e., does not have a telephone plug 167a inserted therein). As a result, if RF port 167 is open, the RF signals from high-pass filter 164 (which is, for example, a single capacitor inserted in series on the red or green wires) pass through switch 165 to terminator 163. Terminator 163 absorbs all of the RF energy transmitting from the network to port 167, allowing no reflection. This can be achieved with a simple resistor (such as approximately 100 ohms) that matches the impedance of the telephone line and connects from the red to the green wires.

When telephone plug 167a is inserted into port 167 (as shown in Fig. 1A), plug 167a pivots arm 165a downward, compressing spring 167b and changing the position of switch 165 to couple the RF signals between high pass filter 164 to RF port 167, bypassing terminator 163.

<u>Frequency Modulation to Decrease Minimum SNR and Reduce Distortion</u>

In the general procedure described in U.S. Patent No. 5,010,399, video signals are converted to RF bands

before being fed to the telephone wiring. Some of the specific conversion techniques described include the modulation of a basebanded video signal to an AM channel tunable by ordinary televisions.

Amplitude modulation (AM) has the advantage of being relatively inexpensive and narrow in bandwidth. Its drawback is that a high SNR is required for good picture quality. For NTSC signals modulated with a one-sided bandwidth of 4 Mhz, an SNR of at least 40dB is required. (The one-sided bandwidth is defined as being the distance from the picture carrier to one end of the band.)

Modulating video signals using frequency modulation (FM) can alleviate the problems of high SNR requirements because the FM reception process is generally more sensitive than AM reception. This advantage follows from the fact that the SNR at the output of an FM receiver is generally higher than the SNR at its input. In other words, the "signal-to-noise" in FM is higher than the "carrier-to-noise." (In AM, by contrast, the "signal-to-noise" is equal to the "carrier-to-noise.")

The improvement in minimum SNR depends on the nature of the noise, the nature of the reception and demodulation process and, in particular, the bandwidth of the signal. All other factors being equal, an improvement in minimum SNR will always accompany an increase in FM bandwidth. One example of the relationship of bandwidth to the SNR improvement is the VFMS-2000 system, an FM video modulate/demodulate pair built by CATEL Corporation. This pair uses 14 Mhz of bandwidth and provides an SNR improvement of approximately 10dB over AM communication.

FM video signals with bandwidths wider than 20 Mhz are used in communication with satellites, resulting in advantages in sensitivity greater than 10dB. As the modulation index and thus the bandwidth increases, however, higher frequencies are required, causing increased attenuation at the high end of the signal, possibly

canceling the extra advantages. Nevertheless, use of very wideband FM signals in transmission of video signals in the applications addressed herein holds the promise of significantly increasing transmission length.

The power of FM signals to reject interference increases when the interfering signal is a second FM signal confined within the same channel. The minimum energy advantage that a receiver requires to reject a weaker but otherwise equivalent signal in the same channel is known as the "capture ratio", and is often significantly less than the minimum SNR necessary to avoid distortion by white noise. The exact capture ratio will depend on several factors, but the inventors estimate that the capture ratio of an FM NTSC video signal with a 15 Mhz bandwidth will typically be less than 10dB, allowing it to ignore interfering FM signals whose levels are suppressed by at least 10dB.

Another advantage of frequency modulation is that it makes the signal less susceptible to spectral tilt. Spectral tilt, which is described in U.S. Patent No. 5,010,399, occurs when the signal energy at one end of a signal spectrum is out of proportion to the energy at the opposite end. When the difference is large it can cause distortion of amplitude modulated (AM) signals because information is carried in the amplitude variations of the signal. Frequency modulated signals, by contrast, are relatively immune to spectral tilt because their information is encoded in frequency variations.

Spectral tilt often occurs during transmission because the attenuation (per unit length) of the medium increases at the high end of the spectrum. The problem increases as transmission length increases. Wideband AM signals, such as standard NTSC video signals, are especially susceptible because the difference in transmission attenuation between the high and low ends of their spectrum is likely to be more pronounced. By

contrast, narrowband (such as audio) AM signals rarely encounter this type of distortion.

In the case of standard amplitude modulated NTSC video signals, the information in the lower half of the frequency band of the signal is redundant and is ignored by receivers. Thus, the difference between the signal level at the picture carrier frequency and the signal level at the upper end of the band (which is 4 Mhz above the carrier) determines whether interference due to spectral tilt is likely to occur.

Compensation for spectral tilt can be implemented at the receive end of the transmission path by boosting the level of the higher frequencies by the amount of extra attenuation that they experienced during transmission. The extra attenuation can be estimated, and the compensation "fined tuned" in response to that estimate. This is called equalization, and requires additional processing which raises costs and adds complexity. Alternatively, the higher frequencies of the signal can be amplified commensurately with the extra attenuation expected during transmission. This is called pre-emphasis and increases cost for the same reasons. If adjustable pre-emphasis or equalization circuitry is provided, the amount of compensation can be "fine tuned" in response to the observed quality of transmission.

Inspection of the relationship between signal frequency and the attenuation of signals by telephone wiring reveals the frequency bands in which the difference in the rate of attenuation between the two ends of a 4 Mhz band is significant. The attenuation of signals transmitting on telephone wiring at 61.25 Mhz, for example, is approximately 11 dB per 100 feet. At 65.25 Mhz, the rate is approximately 11.66 dB per 100 feet. Thus, the low end of an NTSC video signal transmitting at VHF channel 3 (which spans between 60 and 66 Mhz) will gain a .66 dB advantage over the high end for every 100 feet of path

length. Similar analysis shows that the differential across any 4 Mhz channel is approximately the same for 4 Mhz bands whose low end exceeds 5 Mhz. (E.g., attenuation at 20 Mhz is .66 dB less per 100 feet than attenuation at 24 Mhz.) This is not generally true for frequency bands whose low ends are below 5 Mhz. This means that spectral tilt is not an important factor when comparing two bands whose low ends are both above 5 Mhz.

When the ratio of the upper frequency limit to the lower frequency limit of a transmission channel is very large, spectral tilt can cause interference to FM signals if two signals within the same frequency band (channel) transmit on neighboring twisted pairs in a bundle. This problem, and a proposed solution, is described below in the section that addresses the transmission of signals on tightly bundled twisted pairs.

Another advantage of frequency modulation is that it eliminates another form of distortion related to the varying attenuation caused by connected telephones. That type of distortion is described later on in this application.

One drawback of frequency modulation is that it complicates the design of the video receiver. Specifically, RF converter 19 in Fig. 2 of U.S. Patent No. 5,010,399 must convert the waveform of the video signal in addition to converting the signal to a different frequency band. This is because most televisions can only receive AM signals. One preferred method is to detect the FM signal, thereby providing a signal in the baseband frequency range. The basebanded signal is then amplitude modulated to a tunable channel.

Transmission at Empty VHF and
UHF Channels to Reduce Interference

One method of reducing interference is to transmit the video signals within bands that are allocated by the FCC for television transmission but are not being used in the local area. Because the bands allocated for video are

always off-limits to other types of sources, this assures the absence of any broadcast interference. This assures that the IF component in governing equation #1, above, is always a minimum. This makes transmission at these frequencies more reliable from the standpoint that there is no danger that the IF component will have a dramatic and sudden increase due to a nearby broadcast source.

For example, frequencies below 30 Mhz are susceptible to interference from a nearby amateur radio (HAM) transmitter operating in the 10, 15, 20, and 30 meter bands. The probability of such interference is small because the broadcasting antenna must be very close. Where there is no tolerance for such interference, however, the unused television channels are more favorable than the frequencies below 30 Mhz, despite the increased transmission path length.

In U.S. Patent No. 5,010,399, reference is made to the use of empty video channels below VHF channel 7. Channels at VHF 7 and above were not considered good candidates because of the extra radiation that would accompany their higher frequencies. By using frequency modulation or by installing low-pass filters at each telephone (such as by using splitter 161 above), however, the length over which signals at higher frequencies can transmit is significantly increased. Because only two low VHF channels, VHF 3 and VHF 6, are empty in, among others cities, Los Angeles, New York, Chicago, Detroit, and Boston, these high VHF empty channels can be important.

In AM transmission, the signal bandwidth and channel separation match the standard NTSC 6 Mhz channel system, so an AM signal can fit into any unused VHF or UHF television channel. FM transmission, by contrast, loses much of its advantage in minimum SNR when its bandwidth is confined within a 6 Mhz channel. Thus, consecutive empty channels must be used for FM. Unfortunately, many large cities do not have consecutive empty channels in the VHF band. In

New York City, for example, only channels 3, 6, 8, 10, and 12 are unused.

Many consecutive unused channels exist, however, in the UHF band, allowing one to find enough bandwidth to accommodate an FM signal. Although the wire attenuation is higher at UHF frequencies, and radiation from the wire is also higher, if the telephone attenuation is significantly reduced (e.g., by using low-pass filters at each telephone) and the low minimum SNR advantage of FM is exploited, transmission at these channels over the internal residential telephone network may well be commercially feasible.

<u>Eliminating Disturbance of RF Video Signals Caused by Voiceband Energy (Fig. 3)</u>

The inventors have determined that, in addition to the attenuative effects of some telephones connected near the dominant path, certain telephones would occasionally disturb the television picture when voiceband signals were present on the telephone line. The inventors did not correlate this interference with any particular class or category of telephone device.

In approximately one third of the residences tested by the inventors, a disturbance in the displayed picture was observed when any ordinary voiceband signal such as a typical telephone conversation, a dial tone, touch tone, or rotary dial signal, was present. The interference was also noticed when a ring signal was applied to the telephone. Generally, if a voiceband signal caused a disturbance, the ring signal did as well. Conversely, if voiceband signals did not cause a disturbance, ringing signals also did not.

The inventors traced the problem to one or more of the telephone devices connected to the network. Some of these devices only caused their disturbance when off-hook, some only when on-hook, some in either condition. The problems only occurred when these devices were connected close to the dominant path -- as the distance to the

dominant path increased, the interference always lessened. (When a video signal with a 25 MHz carrier was transmitting, telephones ceased to cause a disturbance when removed 60 ft from the dominant path.) When a low-pass filter was interposed between the telephone and the wiring as described above, the disturbance disappeared completely.

Evidence indicated that variations in attenuation of RF energy by such telephone devices closely tracks the variations in the time-varying voltage that represents the voiceband signals. This varying attenuation causes a rapidly varying video signal level at the video receiver. If the telephone device that induces the varying attenuation is connected close to the dominant path and the variations are large, the interference will be significant. As is seen from the discussion above, amplitude modulated signals are much more likely to be affected by this interference than frequency modulated signals.

One method of substantially eliminating this problem in a given residence is to install low-pass filters on every telephone. This was suggested earlier to expand the number of transmission channels. It is certainly feasible when installation of a video transmission system is performed professionally. It may not be practical, however, to require an ordinary consumer to perform this installation.

An alternative solution is to install an automatic gain control (AGC) circuit in the RF device that receives the video signal (e.g., the television transceiver shown in Fig. 2 of U.S. Patent No. 5,010,399). The AGC circuit smooths out the variations in video signal level caused by offending telephones before presenting that signal to the television. A description of a circuit that can perform this function is given below.

Let the attenuation of the offending telephone be represented by the following equation (#2):

$$C + Bi(t)$$

where C represents the constant component of the attenuation and Bi(t) represents the time varying portion of the attenuation. B is a constant and represents the magnitude of the variations, and i(t) is zero mean, unit power and represents the variation with time.

Now let the video component of the television signal at the receiver when the telephone is disconnected be denoted by Vv(t), where V is a constant and v(t) has a peak of unity and zero mean. The received signal when the phone connects is thus:

$$[C + Bi(t)]\ Vv(t) \qquad \text{equation (\#3)}$$

or

$$CVv(t) + VB\ [i(t)v(t)] \qquad \text{equation (\#4)}$$

or

$$[1 + (B/C)i(t)]\ CVv(t) \qquad \text{equation (\#5)}$$

The first term in equation #4 represents the video signal and the second term represents the noise.

Note that the quantity [C + Bi(t)] in equation #3 (and the mathematically identical quantity [1 + (B/C)i(t)]C in equation #5) multiplies the pure video signal Vv(t). Thus, this quantity represents what can be called "multiplicative noise." This quantity is time varying due to the process i(t), described herein. This quantity is also known as the "envelope." By smoothing out this variation, i.e. by using an AGC (automatic gain control) circuit in the receiver to apply a time varying gain equal to the inverse of this quantity, the noise can be canceled and only the signal, v(t), will remain.

Applying AGC techniques to AM video, unfortunately, presents an additional difficulty. The difficulty lies in the fact that amplitude variations due to interference are not easily distinguishable from variations that represent the modulated signal. The solution disclosed herein (and shown in Fig. 3) measures variations in the amplitude of the sound component of the video signal to estimate the behavior of the interfering signal. This is possible

because the sound carrier is frequency modulated, meaning that variations in its amplitude do not represent signal information, only interference. This interference, furthermore, will be very similar to the signal interfering with the video component. That is because the video and sound components are relatively close in frequency. (The picture carrier in a NTSC signal is separated from the sound carrier by only 4.5 MHz.) It follows that the interference measured in this manner can be used to compensate for the corruption of the video component. This procedure is described below.

Let Ss(t) represent the sound component of the television signals, where s is constant. Because this signal is frequency modulated, the quantity s(t) is a sinusoid with time varying frequency. If that sinusoid is assigned an amplitude of 1, S becomes the amplitude of the signal.

When Ss(t) replaces Vv(t) in equation 5, the resulting expression, [1 + (B/C)i(t)]CSs(t), represents the disturbed sound signal. Furthermore, [1 + (B/C)i(t)]CS can be viewed as the time varying amplitude of the sinusoid s(t), because i(t) varies much more slowly (it varies at voiceband frequencies) than s(t). Because the inverse of the quantity k[1 + (B/C)i(t)]CS (where k is a multiplicative constant) when multiplied by the video signal [1 + (B/C)i(t)]CVv(t) leaves the pure video signal (k/S)Vv(t), if k[1 + (B/C)i(t)]CS can be estimated, the video interference can be canceled.

An estimate of the time varying amplitude of the sound signal, [1 + (B/C)i(t)]CS, is computed by computing the RMS of that signal over an averaging time long enough to smooth out variations in s(t), but short enough to preserve variations of i(t). Thus, the lower bound of the averaging time will be the inverse of the highest frequency of v(t), i.e. a value in the microsecond range. An upper bound will be the inverse of the maximum frequency of the

baseband signal i(t). This value will be in the millisecond range because i(t) is in the telephone voiceband. An estimate of [1 + (B/C)i(t)]CS will be .707 times the computed RMS.

Fig. 3 shows a block diagram that illustrates the estimation process. This process can be employed in, e.g., television transceiver 15 of Fig. 2 in U.S. Patent No. 5,010,399. A television signal is split and fed to bandpass filter 300 and gain control 303. The filter attenuates the video component, leaving only the sound component, which is fed to RMS circuitry 301. That component estimates the RMS of the sound signal over an averaging period that is set according to the above description. As described above, this represents the multiplicative noise in the television signal. This time varying quantity is inverted by invertor 302, and the resulting signal is used to control the gain applied by gain control 303 to the television signal. Varying the gain in this manner removes the noise according to the procedure described above.

Standard television gain control circuits monitor the energy of the video signal, and apply an amount of attenuation or gain necessary to keep the signal at a desired level. Thus, the gain control smooths out the variations in the amplitude of the received signal.

Standard gain control circuits have a response time of seconds. The amplitude changes caused that invertor 302 instructs gain control 303 to implement, however, occur at the rate of the highest frequency of the voiceband signals, i.e. 5 KHz. This requires gain control circuits to react at least this fast, i.e. .2 milliseconds, in order to track voiceband changes effectively. This reaction rate is higher than that of gain control circuits typically used for video signals, but not beyond the most rapid circuits that can be built with inexpensive electronics.

As discussed above, frequency modulated video

signals are more immune to interference and are less likely to be disturbed by the phenomenon described in this section. In the event that this interference does corrupt an FM video signal, however, the compensation process can be simpler than the procedure disclosed above. This is because the amplitude of the FM video signals, like FM sound signals, is not varied with time by the modulation, meaning that the amplitude variations of the video component correspond to the interfering signal. Thus, an AGC circuit treating an FM video signal can react to variations in the video amplitude directly, rather than the variations in the sound carrier, and conduct the smoothing operation in the ordinary manner.

<u>Transmitting RF Signals over Two Different Wire Pairs in the Same Bundle</u>

As discussed above, normal internal telephone wiring includes four conductors. Voiceband signals typically use the red/green pair for the first telephone line, and use the yellow/black pair if a second line is connected. Some wiring includes many pairs within the same bundle (i.e., enclosed within a single sheath).

Some of the energy of RF signals can cross over from one wire pair to an adjacent pair within the same bundle, especially on four conductor wire. As frequency increases, this crosstalk effect becomes larger. This will cause interference and prevent the use of the same frequency to transmit different signals on separate pairs in the same bundle. The crosstalk effect thus limits the opportunities presented by extra conductors to the lower frequency ranges. An example is a cable consisting of a bundle of telephone wire pairs, and whose properties are such that when energy is fed onto one pair at 20 Mhz, it can be received, through crossover, at the end of the cable on a neighboring pair at a level only 40 dB lower than the level on the identical pair. Because AM NTSC signals have a minimum SNR requirement of at least 40dB, this means that different signals cannot be transmitted onto different

pairs at frequencies above 20 Mhz.

If two signals at the same frequencies are fed onto different pairs at separate locations on a network, the interference will depend on the relative locations of the transmitters and receivers as well as the crosstalk. A more interesting and important question is whether two signals fed onto different pairs at the same point on the internal telephone network will interfere. This type of interference is called "far end crosstalk."

Because they are at the same frequency, the energy level of these two signals will decrease at the same rate. Thus, the levels reaching their respective receivers or reaching the point where the pairs separate, will be nearly the same. Also, the amount of energy crossing from one pair to the second will approximately equal the energy crossing in the reverse direction. Furthermore, if the crosstalk energy is higher than other noise energy at the receiver, the SNR seen by either receiver is the ratio of the energy of the signal of interest to the energy crossing over from the neighboring pair. The ratio of signal to noise in this case is simply:

$$SNR = SL1 - (SL2 - CR) \quad \text{equation (\#5)}$$

where SL1 is the source level of the signal of interest, SL2 is the source level of the signal on the other wire pair, and CR is the loss suffered by SL2 in crossing over. Because SL1 = SL2, the SNR is simply CR. If this is less than the minimum SNR for the signal, the crosstalk effects will not degrade the video signal displayed by the television. The quantity CR is called the "far end crosstalk loss."

Because the minimum SNR of AM video signals is at least 40dB, even a small amount of crosstalk can cause noticeable interference in the television picture. Because FM video signals have a capture ratio of less than 10dB, however, the possibility that the second pair can provide extra video channels is significantly higher when FM is used.

A problem can occur, however, when the ratio of the upper to lower frequency limits of the transmission band is large, i.e., when the spectral tilt is large. The problem occurs during the instant of time when the carrier of the signal of interest is at a high frequency deviation while the carrier of the interfering signals is at a low deviation. Ordinarily, if the crosstalk loss is greater than the "capture ratio," interference will not occur. Because attenuation at the higher end of the band can be dramatically higher than that at the low end, however, the energy of the interfering signal can actually be greater than that of the signal of interest.

For example, assume two signals are frequency modulated between 10 Mhz and 60 Mhz, and are transmitted onto different twisted pairs within a bundle 500 feet long. Attenuation at 60 Mhz is approximately 10dB per 100 feet, while attenuation at 10 Mhz is approximately 3 dB per 100 feet. After a transmission distance of 500 feet, therefore, the interfering signal when it is at 10 Mhz will be 35dB higher than the signal of interest when it is at 60 Mhz. Thus, if the far end crosstalk loss is less than 35dB, the interfering signal will be at a higher level, and the SNR will be less than 1.

The solution proposed herein is to apply the equalization or pre-emphasis process described above to frequency modulated signals. In that way, the received signal levels will be equal across frequency, and the interfering signal will not have a relative advantage when it is at lower frequencies. In the specific example given, pre-emphasis would provide the signal energy that is at 60 Mhz at a level 35dB higher than the energy at 10 Mhz. In that way, the levels of both frequencies at the receive end would be similar.

Transmission of High-Fidelity Audio Signals Across Telephone Wiring (Figs. 4A-4B)

Because most of the energy generated by high-fidelity audio systems is concentrated in the voiceband, signals from those systems will interfere with telephone communications when transmitted across telephone lines. The solution disclosed herein uses a concept similar to that described in U.S. Patent No. 5,010,399 for transmitting infrared signals across active telephone wires. Signals transmitted using that technique are first converted to a higher frequency band, then amplified before transmission onto the wiring. The resulting signal is received at the end of a path, and used to recreate the original waveform at baseband.

The application of this method to high-fidelity audio signals is shown in Fig. 4A. Left and right stereo channels at pre-amplified levels are passed from a sound system 151 to hi-fi transmitter 150. Modulator 152a modulates the left channel at a first RF carrier frequency, (e.g., 45 MHz) and the right channel is modulated at a different RF carrier frequency, (such as 50 Mhz) by modulator 152b. Different carrier frequencies are used so that the modulated signals do not interfere with each other when they are combined by coupler 153 onto the same conductive path. Because well-respected consumer electronic standards establish consistency in the voltage of pre-amplified signals, design of modulators 152 can achieve an economy by relying upon input levels within a narrow amplitude range.

The carrier frequencies must be high enough to convert all of the signal energy above voiceband. It may also help to leave the signals within a band where less governmental restrictions apply. In the U.S., for example, the Federal Communications Commission does not allow any energy below 270 Khz to be fed to the public telephone network. They do allow, however, levels of -30dbV above that frequency. The U.S. FCC places no limits at all on

energy above 6 MHz in frequency.

The typical method for modulating audio signals is to use techniques such as AM, FM, or SSB (single sideband). Each of these methods, of course, includes a companion demodulator which converts signals back to their original form. A modulator/demodulator pair that cooperate in this manner may be thought of as a simple radio station and radio receiver that use the telephone wiring as a transmission medium.

FM transmission is the preferred method because the fidelity of a signal transmitted using that technique is higher than if AM or SSB were used with equally expensive circuitry. Signals converted via frequency modulation also have the added benefit of greater immunity to interference. The audio quality when using FM would be commensurate with standard FM stereo reception. It could even be improved by using higher quality modulation circuits, or by increasing the bandwidth beyond the FCC regulations which restrict the bandwidth of broadcast FM. (More bandwidth is available on the telephone lines because the only frequencies that are occupied on that medium are voiceband frequencies. Also, the bandwidth of FM broadcast stations is approximately 150 KHz, meaning that there is plenty of spectral space available for these types of signals, even if their bandwidth is more than doubled.)

Coupler 154 applies the modulated signals to amplifier 154, and the amplified signals are passed through bandpass filter 155 to coupling network 156. Filter 155 restricts passage of energy between amplifier 154 and coupler 156 to the frequency bands occupied by the modulated hi-fi signals. This prevents extraneous signal output from amp 154 from exiting onto the active residential telephone network 160 and prevents amp 154 from loading down RF signals that may be coupled across network 160 at different frequencies.

Coupling network 156 is shown in Fig. 4A. It

includes hi-pass filter 146, and balancing and impedance matching circuitry 175. RF signals passing to coupling network 156 from network 160 pass through hi-pass filter 146, which blocks low-frequency (e.g., voiceband) signals on network 160 so that connection and operation of transmitter 150 does not disturb telephone communications. Filter 146 can be implemented by a single capacitor connected in series along either of the two wires. It is not needed if the telephone wiring is inactive. The RF signals then pass onto network 160.

Balancing and impedance matching circuitry 175 matches the impedance of the telephone line, reducing the energy radiated by RF signals crossing that junction, and increasing the efficiency of transmission onto the wiring. It also balances the voltage of signals transmitting in the opposite direction, (i.e. onto the telephone network.) This also reduces radiation of energy. Balancing and impedance matching circuitry are shown in Figs. 6 and 7 of U.S. Patent No. 5,010,399, for a coupling network that served as a junction of three paths. Those skilled in the art can convert the wound-torroid described therein to achieve the balancing and impedance matching results for this case, which is a junction of two paths.

Transmitter 150 also includes low pass filter 158a and port 159 to allow connection of telephone devices 145 to network 160 through transmitter 150. Filter 158a isolates telephone devices 145 from network 160 at high frequencies, preventing devices 145 from loading down the modulated signals transmitted by coupling network 156. Filter 158a can also constitute a component separate from transmitter 150. For example, low pass filtering is used to connect other telephone devices 145 elsewhere on network 160 (only one such connection is shown). One convenient way of providing the low pass filtering is to connect each telephone device 145 to network 160 with splitter 161 (Fig. 1); in this case, the internal filters 162a, 162b of

spitter 161 provide the low pass filtering.

Once they are applied to internal telephone wiring 160, the modulated signals flow to all points of the network. The hi-fi receiver 170 is connected at any location on network 160 to recover those signals. The recovered signals first pass through coupling network 171. The functional block diagram for coupling network 171 is the same as that for coupling network 156 (Fig. 4A). Signals are received through a high pass filter that presents a high impedance to voiceband signals, preventing the connection of receiver 170 from disrupting telephone communications. (This filter is not needed if the wiring is not active.) The signals next encounter balancing and impedance matching circuitry (similar to that discussed above) to match the impedance of the telephone wiring to the impedance internal to the circuitry of receiver 170. The balancing circuitry unbalances the signal so that it is expressed inside receiver 170 as a voltage relative to ground. The signals then pass through bandpass filter 172, which filters energy outside of the band occupied by the signals of interest. Demodulate and separate circuitry 173 then demodulates each of two signals independently, using known techniques to recreate the two original left and right channel audio signals, which are fed out through ports 174. Demodulate and separate circuitry 173 also adjusts the energy level and impedance of the demodulated signals so that they adhere to the "line out" standards established for audio equipment. Typically, an amplifier (not shown) will be connected to ports 174 to boost the audio signals and drive loudspeakers (also not shown). Of course, such an amplifier can also be provided internally, within the same housing as receiver 170. If an amplifier is provided internally, one need only provide hi-fi receiver 170 and any ordinary pair of loudspeakers to produce the sound signal from sound system 151 at a remote location.

A block diagram of demodulate and separate circuitry 173 is shown in Fig. 4B. Signals fed from filter 172 are split, passing to both filter 146a and 146b. Filter 146a passes only the frequencies of the left channel signal (in this example, 45 MHz), and filter 146b passes only frequencies occupied by the right channel signal (50 MHz). The left channel signal is then processed by FM demodulator 147a, gain control 148a, and impedance matcher 149a. The right channel is processed by identical components.

FM demodulators 147a, 147b demodulate FM encoded signals that occupy the frequencies used by the left and right channel signals, respectively. This demodulation function is well known. After demodulation, the levels of the left and right channel signals are adjusted by gain controllers 148a, 148b to adhere to the well respected standards used for the "line in" and "line out" ports on common audio equipment. Finally, impedance matchers 149a, 149b match the impedance of the conductive paths to the 75 ohm impedance required by the "line out" standards.

Receiver 170 includes low-pass filter 158c and port 176 for connection of telephone equipment. Filter 158c provides the same function as filter 158a. Filter 170 can also be a provided as a separate component.

Fortunately, experiments indicate that internal telephone wiring media are not likely to impose multipath or other distortions as FM encoded audio signals cross network 160. In those experiments, described in U.S. Patent No. 5,010,399, sound signals were transmitted using frequency modulation with center frequencies of 29.75 Mhz and at 65.75 MHz. Those frequencies were the sound carrier frequencies of the NTSC television signals that were transmitted across residential wiring networks.

The FM sound components of those signals were fed onto the wiring at levels of approximately 25dBmV, which was 15dB below the level of the video components. They communicated across all residences without substantial

distortion or degradation. (Degradation of the video, but not the audio component was noticed in approximately 5% of all residences.) The attenuation caused by connected telephones, splits in the wiring, and the wiring itself did not reduce the SNR of the signals enough to affect the resultant sound quality. This is due to the fact that FM receivers can tolerate low SNRs at their inputs without displaying significant interference at their outputs.

Besides revealing that the attenuative influence of the network does not reduce the levels of FM signals enough to cause audible degradation, none of the experiments described in U.S. Patent No. 5,010,399 demonstrated interference from "airborne" RF signals picked up by the wiring. This is partly due to the fact that internal telephone wiring acts as a poor antenna at the relatively low frequencies at which the FM encoded signals are transmitted over network 160, and also because quality reception of FM encoded signals is possible at low SNR levels. Furthermore, because sound signals are relatively narrow in bandwidth, it is easy to find bands that are sufficiently wide yet are not likely to be shared by interfering broadcast energy picked up by the wiring.

In some residences and most small offices, telephone networks consist of several dedicated paths that connect directly to a central switch, sometimes called a PBX for private branch exchange, or KSU for Key Service Unit. The conductive paths across this network are usually broken by such a switch. Such a break poses a barrier to the communication of video signals, as is described in U.S. Patent No. 5,010,399. The same problem will be encountered by audio signals transmitted using the techniques described herein.

Adapter 52 shown in Fig. 5 of U.S. Patent No. 5,010,399, when installed at such a network switch isolates that switch from RF video signals while allowing those signals to flow freely from one path to another. An

adapter designed on the same principles will do the same thing for RF audio signals. In the event that the dedicated paths in a PBX network are very long, the attenuation of the wiring can cause the SNR at the receiver to fall below acceptable levels. As described in U.S. Patent No. 5,010,399, an amplifier can be added to the adapter to boost the signal level before the second leg of the transmission path is traversed.

An interesting variation on the system of Fig. 4a is to encode the left and right hi-fi channels using the same modulation system that FM radio stations use to broadcast stereo signals. When this is done, an ordinary radio receiver can receive the signals by connecting its antenna terminals to the telephone wiring through a high-pass filter. Replacing demodulate and separate circuitry 173 in device 170 by an antenna connected to a FM stereo radio provides that result.

<u>Transmission of Digital Signals Across Telephone Networks (Fig. 5)</u>

Transmission of high data rate digital data streams across internal telephone wiring can be accomplished with commercially available devices known to transmit across that medium using wideband signals and frequencies above voiceband. Some of these devices allow communication across wiring that is conducting voice communication. These devices are sometimes used in offices as part of computer communication networks.

These communication systems always transmit signals from one point to another along a "point-to-point" wire that includes no splits or other junctions. An open question is whether these devices can achieve transmission over telephone wiring that is not "point-to-point" but includes many randomly connected paths. This would allow digital devices to communicate in a broadcast fashion, where a signal fed onto the wiring by a digital source spreads across the entire network and is available to a receiver that is connected at any branch of the network.

The method disclosed herein for transmission of digital signals is based on the same principles as those described above and in U.S. Patent No. 5,010,399 for video and hi-fi signals. Experiments described in U.S. Patent No. 5,010,399 demonstrated that NTSC video signals can transmit across networks of residential telephone wiring without substantial distortion. Because very high data rate digital signals have a bandwidth similar to that of NTSC and can often be received with lower SNRs, those experiments indicate that data signals can also transmit in this manner.

Referring to Fig. 5, digital transmitter 178 is disclosed to feed high data rate digital signals, such as the 19.2 Kbit/sec signals generated under the IEEE RS-232 standard, onto active internal telephone network 188. Following is a description of the steps in the transmission process:

1) Digital signals are derived from a digital source 180, such as a the serial port on an IBM compatible PC.

2) The digital signals are fed to signal conditioner 181 that transforms the high and low voltages on the various conductors or "pins" of the port into a single analog wave expressed as voltage variations. This waveform may be as simple as a bi-level signal. It embodies not only the basic signal, but also information required for coordination with the receiver. The output signal produced by conditioner 181 is in the form at which the signal can be efficiently transmitted across telephone network 188.

Many techniques are known to perform this conversion, such as the Bell 212 standard, which uses "phase-shift-keying" to achieve 1200 band communication in common modems. Devices that transmit signals according to the Bell 212 standard can input data from the serial port of a PC, and feed an analog waveform at voiceband frequencies onto an active telephone line. Another example of this type of conversion is a technique known as Manchester coding, which outputs bi-level waveforms.

3) After being expressed as a voltage variation, the signal may be shifted to a different

frequency using known heterodyne techniques. This is accomplished by RF converter 182. If some of the energy of the signal output from conditioner 181 is within the voiceband or overlaps the band used by other signals, this process is required. Otherwise, the process is useful only to take advantage of the different properties of the wiring at the shifted frequency.

Some conditioning techniques convert digital signals to the form of square waves whose energy is concentrated at frequencies above voiceband. Examples are the transmitters of LAN (Local Area Networks) that adhere to the 10BaseT standard. If conditioner 181 outputs its signal in this manner, converter 182 will not be required.

4) The level of the signal is increased by amplifier 185, and the amplified signal is coupled through a bandpass filter 183, which blocks energy outside the band confining the signal (i.e., voiceband). The signal now occupies the frequency channel at which it will transmit across telephone wiring 188.

5) The signals are then fed through a coupling network 184 and on to the network wiring 188. Network 184 balances the signal and matches the impedance of the telephone line. Network 184 also includes a high-pass filter on the port connecting to the telephone wiring. That filter blocks voiceband energy, making connection and operation of transmitter 178 transparent to voiceband communication. The requirements of coupling network 184 are the same as the requirements of coupling network 156, shown in Fig. 4a.

6) Port 184a is provided for connection of telephone devices. This port connects to the wiring through low-pass filter 184b that prevents those devices from draining the RF energy.

Signals transmitted according to the above process will ordinarily transmit across the entire network 188, and will thus be available to any cooperating receiver 179 that connects to the wiring anywhere in the residence.

Digital receiver 179 is also shown in Fig. 4. Following are the steps in the receiver process:

1) Receiver 179 connects to network wiring 188 to derive signals transmitting across that medium.

2) The signals are fed through coupling network 189, which performs the same functions as coupling network 171. The signals first pass

through the high-pass filter of that network. That filter blocks voiceband energy. Next, coupling network 189 matches the impedance of the telephone line and unbalances the signal.

3) Signals emerging from the coupling network pass through band-pass filter 193, which attenuates energy outside the passband of the signal.

4) The signals are then applied to signal processor 190, which converts the signals to baseband frequency with an energy level inside the range expected by signal conditioner 191. This conversion may involve a shift in frequency or a demodulation, each of which can be accomplished using well known techniques, and are the inverse of the treatment provided by RF converter 182. Processor 190 may also perform an alteration of signal level using known AGC (automatic gain control) techniques. This is necessary because the level of the signal fed to the line by transmitter 178 may be very high, and if the transmission path is short, the signal received by receiver 179 will also be very high. An AGC can reduce the level of this signal to a range that is more easily managed by ordinary electronics. If transmitter 178 does not include RF converter 182, and the level of the signal received at network 189 will always fall within the range permitted by conditioner 191, processor 190 will not be required.

5) Signal conditioner 191 converts the voltage variations output by processor 190 into a digital data stream in a form expected by the connected digital terminal device 192. When the output of processor 190 is a square wave, digital devices may be able to read this output directly. In this case, conditioner 191 is not needed.

6) Port 189a is provided for connection of telephone devices. This port connects to the wiring through low-pass filter 189b that prevents those devices from draining the RF energy.

Techniques are disclosed herein and in U.S. Patent No. 5,010,399 to increase the maximum path length of transmission of video signals. These techniques will also facilitate transmission of high data rate digital signals as described above. Following is a partial list:

1) providing each telephone port on the network with either a low pass filter (shown in Fig. 5 as LPF 188a), or splitter 161, which includes a low-pass filter.

2) using frequency modulation in converter 182

to increase interference rejection;

3) choosing transmission channels that are less likely to be used nearby broadcast sources, thus reducing the chance of interference;

4) reducing transmission frequency to lower the attenuation caused by the wiring, to lessen the ability of the wiring to pick up interference, and to allow higher signal levels without violating airborne radiation regulations;

5) providing a low-pass filter, shown in Fig. 5 as filter 188b, along the path connecting the network to an external signal source, such as the public telephone system, in order to suppress the higher harmonics of ringing voltage and switch-hook transients originating at the external source.

Transmitter/receiver pair 178, 179 can also achieve two-way communication by transmitting data in the reverse direction, from receiver 179 to transmitter 178, over the same pair of telephone wires of network 188 (but over a different frequency band) using the same techniques as those described above. Techniques for simultaneous transmission and reception of various signals through a single connection to the wiring are disclosed in a later section of this document (entitled "Simultaneous Transmission of Multiple RF Signals Across Internal Telephone Wiring").

Transmitter/receiver pair 178, 179 can also use the same channel for alternating two-way transmission if they cooperate to ensure that only one device is actively transmitting at any one time. Such systems of cooperation are used in well-known computer communication networks.

Because the digital transmission technique described above is independent of the type of information represented by the data streams, digitized video signals can transmit across networks of telephone wiring using that method. Transmission of digital video using this technique is facilitated by advancements in the compression of digitized video signals. These have enabled an impressive reduction of the data rate of the signal bitstreams and, consequentially, an impressive reduction of the bandwidth

required to transmit the signal. Commercial quality video signals can now be represented by analog waveforms covering less than 2 Mhz of spectrum.

When transmitter 178 and receiver 179 transmit digital video using the techniques described above, the process is a specific embodiment of the more general technique disclosed in U.S. Patent No. 5,010,399. The functions performed by RF converter 3 of U.S. Patent No. 5,010,399 correspond to those performed by signal conditioner 181 and RF converter 182, and those performed by RF converter 19 in transceiver 15 in U.S. Patent No. 5,010,399 correspond to the functions performed by signal processor 190 and signal conditioner 191. The amplifiers, bandpass filters, and coupling networks of the corresponding devices also perform identical functions.

Transmission of Hi-Fidelity Audio Signals
Across Telephone Networks
Using Digital Techniques (Fig. 6)

A system for transmitting high-fidelity audio signals based on FM techniques was described earlier in this document with reference to Figs. 4A-4B. Inexpensive electronic components that perform FM modulation, however, may not be precise enough to support the sound quality generated by audio components that operate on digital principles. It is for this reason that, for example, music created directly from compact discs meets higher specifications than music received from FM broadcasting, even if the source of the broadcast music is a compact disc.

Digital transmission techniques provide an acceptable alternative. The proposed procedure (shown in Fig. 6 and discussed in detail below) begins by digitizing audio signals or starting with a digital audio source. These signals are transmitted using the techniques described in the previous section. The analog signal is then reconstructed at the receive end. Digitizing and reconstructing can be accomplished by devices known to

digitize sound signals and to transform them back with no significant loss in quality.

The digital transmission concept is shown in Fig. 6. The system includes hi-fi transmitter 200 that accepts analog pre-amplified left and right channels from an analog stereo system (not shown) or digital audio channels from a digital stereo source 204. Transmitter 200 processes these signals and transmits them onto the active telephone wiring 220. Hi-fi receiver 210 recovers these signals from the telephone wires at a second location on network 220 and converts them to their original, pre-amplified form so that the audio signals can be used as input to a speaker/amplifier system.

Transmitter 200 accepts the left and right channel analog stereo signals at ports 201 and 202 and transmits them to digitize and compress circuitry 203a and 203b, respectively. Because well-known consumer electronic standards establish consistency in the voltage of pre-amplified signals, design of circuitry 203a and 203b can achieve an economy by relying upon input levels within a narrow amplitude range.

According to mathematical principles, the digitization rate must be at least twice as high as the highest signal frequency in order to capture all of the information. Thus, 50,000 samples per second will capture all information up to 25,000 Hz, a frequency slightly higher than the highest frequency used in standard digital sound systems, and above the range of human hearing.

The left and right channel analog signals are digitized and compressed by converter and compression circuitry 203a, 203b, respectively. The preferred method is to use the standard digitization and compression procedure used to create common compact discs. The advantage of that method is that inexpensive integrated circuits are available to accomplish digitization and compression according to that standard. Use of a CD coding

system for circuitry 203a, 203b also ensures that the transmission process will maintain CD sound quality within the system.

The CD system uses 16 bits to represent each sample of the signal, and uses compression techniques to reduce this quantity to 12 bits. Because 50,000 samples encoded at 12 bits each results in 600,000 bits, digitize and compress circuitry 203a, 203b will each produce a datastream of 600,000 bits per second.

Some hi-fi components, especially CD players, output their signals as digital datastreams as well as in analog form. When connecting to these players, circuitry 203 is not necessary. Port 204a is provided to receive these digital outputs and to feed them directly to RF encoder 205, the next step in the processing stream.

To transmit this digital information across telephone wiring at a very low error rate, known circuits common to computer "local area networks" can be used. The two datastreams are passed to this type of circuit, RF encoder 205, which represents each of them as variations of voltage across two wires at a frequency above voiceband. The input to RF encoder 205, by contrast, is digital and is typically expressed as time varying bi-level voltages on several conductors. An example of an RF encoder that inputs digital signals and outputs an RF signal between 3 MHz and 15 Mhz, i.e. above the voiceband, are any of the transmitters that adhere to the IEEE 10BaseT standard. (As described in the background section of this document, that standard governs the Local Area Networks (LANs) that transmit 10M bits/sec of data over twisted pair wires that are dedicated for point-to-point communication.)

The signal generated by encoder 205 passes through coupling network 206 onto telephone network 220. Network 206 feeds that signal to telephone wiring 220 through a hi-pass filter that prevents disturbance with telephone communications. (This hi-pass filter is not necessary if

the wiring is inactive.) Coupling network 206 also matches the impedance of the telephone line and balances the signal across the two leads of the telephone wiring. This reduces radiation and increases the efficiency of transmission onto the wiring. The function of coupling network 206 is identical to that of network 156, shown in Fig. 4A.

Transmitter 200 also includes a low-pass filter 216 and a port 215 for connection of telephone equipment to network 220. Filter 216 prevents the telephone equipment from loading down the RF signals fed onto telephone wiring 220. To prevent telephone devices connected to network 220 from loading down RF signals from transmitter 200, low pass filters are provided for each telephone. These are shown in Fig. 6 by filter 216a and splitter 161, which includes a low-pass filter. Splitter 161 also provides other benefits when transmitting RF signals across telephone networks. These were described earlier in this document.

The frequency and level of the signal that is fed to telephone wiring 220 is determined by RF encoder 205. As in the case of FM modulated audio signals described earlier, these values should be such that the SNR at the receiving locations is sufficient to provide high quality stereo. In this case, that requirement is roughly equivalent to the requirement of error-free reception of the digital data stream. The signal level must also be low enough to keep RF radiation from the wiring below the legal limits established for the frequencies of the signal, and below the limits on the amount of energy that can be fed to the public telephone network. Experiments performed by the inventors in transmitting video signals across all but the largest residences, indicate that the same combinations of frequency and signal levels, which are within legal limits, will transmit hi-quality stereo over the active telephone lines within all but the largest residences in the U.S. This is because the SNR required at the input to an AM video receiver is much higher than the SNR required at the

input to high data rate digital receivers such as those that adhere to the 10BaseT standard described earlier.

The RF signal is transmitted onto, e.g., a red-green wire pair of telephone network 220 and propagates over the telephone link. At the receiving end, the RF signal is recovered by coupling network 207. The functions performed by network 207 are identical to those performed by network 171 of Fig. 4A. Coupling network 207 feeds the RF signal to RF decoder 208, the companion to RF encoder 205. Decoder 208 recreates the left and right digital datastreams from the recovered signal using known means. Thus the outputs of decoder 208 will typically be time varying bi-level voltages adhering to one of the common standards for digital communications.

The remaining step in the receiving process is to recreate the analog left and right channel audio signals from the digital datastreams. This is the inverse of the digitize-and-compress process performed by circuitry 203, which follows the standard of common CD players and is described above. It is performed by decompress and D/A (digital to analog) integrated circuitry 211a, 211b. The resultant left channel audio signal in analog form is applied to output port 212, and the right channel audio signal is coupled to output port 213. Because the preferred decompress and digital-to-analog circuitry is common to virtually all CD common players, circuitry 211a, 211b can be provided inexpensively.

To recreate the sound, an amplifier (not shown) can accept the recreated signals from ports 212 and 213 and drive speakers (also not shown), which serve as the "receivers" for the audio system. An amplifier can also be provided internal to receiver 210. In this case one need only provide receiver 210 and any ordinary pair of loudspeakers to produce the sound signal from a sound system at a remote location.

Receiver 210 also includes a low-pass filter 209 and

a port 214 for connection of telephone equipment to the network. Filter 209 prevents the telephone equipment from loading down the RF signals fed onto the wiring. Likewise, telephone equipment connected elsewhere on network 220 should use low pass filters, or even more preferably, splitter 161 (Fig. 1).

Because of the limitations, described earlier, of transmitting hi-fi signals across AC power lines, or broadcasting hi-fi signals using radio waves, transmitter/receiver pair 200, 210 constitutes a significant advance in consumer electronics. The advance is even more pronounced when the ability to transmit infrared control signals in the reverse direction is included. Technology to achieve simultaneous transmission of these signals is disclosed in the next section with regard to Fig. 7.

Simultaneous Transmission of Multiple RF Signals Across Internal Telephone Wiring (Figs. 7-9)

U.S. Patent No. 5,010,399, together with the previous sections in this application, describe various techniques for transmitting audio, video, digital, and control signals from infrared transmitters over active networks of internal telephone wiring using radio frequencies. U.S. Patent No. 5,010,399 also describes a pair of transceivers that cooperate to transmit video from one transceiver to the second, and control signals from the second transceiver to the first. In this section, these results are extended to disclose a transceiver that can connect to an the active wiring of a residence to transmit several RF signals of varying types while receiving several others at the same time.

The general design will be described using a pair of transceivers that cooperate to transmit hi-fi, video, and control signals from infrared transmitters across telephone wiring. The processing and signal flow within this pair is shown in Fig. 7.

A video source 251 and a hi-fi source 252 are shown

connected to the transceiver on the left, herein referred to as the video/hi-fi transmitter 250. A television receiver 275 and a speaker/amplifier 276, are shown connected to the transceiver on the right, herein referred to as the video/hi-fi receiver 280.

A wireless infrared source 277, which is normally a hand-held infrared transmitter, sends infrared control signals through the air to video/hi-fi receiver 280. An infrared receiver 253, which corresponds to the infrared pickups on video source 251 and hi-fi source 252, receives the infrared control signal after transmission over active telephone wiring and reconstruction by transmitter 250 (as described in detail in U.S. Patent No. 5,010,399 and as also described below). Telephone equipment 263 and 266 is shown at both ends because it is likely to be connected at any telephone jack. Likewise, telephone equipment 278 is connected through low pass filter 279 (which may be one of the low pass filters of splitter 161 of Fig. 1) at any location on telephone network 264.

Communication of signals across telephone wiring 264 by transmitter 250 and receiver 280 functions as follows. The signals from sources 251, 252 are first processed by respective processors 254, 255 to convert them to the form in which the signals will be efficiently transmitted over the network 264. (The details of this processing is described later on.) These signals are then passed through respective bandpass filters 257, 258 and are combined by a coupling network 260 for transmission over a single pair of wires (e.g., the red-green pair) of network 264. It will be appreciated that coupling network 260 receives signals from all sources connected to transmitter 250, as well as incoming signals recovered from telephone network 264. The combined, outgoing signals emerge from coupling network 260 and pass through a high-pass filter 261 onto network 264. Filter 261 presents a high impedance to telephone signals from network 264 and makes the connection of transmitter

250 transparent to voiceband activity on network 264.

At receiver 280, signals are recovered from the line through a high-pass filter 265 and are applied to coupling network 267. The functions of filter 265 are similar to those of filter 261. Coupling network 267 has an output port for each of the connected receivers, e.g., television 275, and amplifier 276, which are the final destinations of the video and audio signals, respectively. The output ports of coupling network 267 are applied to bandpass filters 269, 270 and then to processors 272, 273 where the video and audio signals are converted to a form compatible with their associated receiver 275, 276, respectively. The details of this processing are described later on.

The general procedure just described is embodied in transceivers 1 and 15 of U.S. Patent No. 5,010,399, which cooperate to transmit video and infrared signals. This procedure is also used to communicate hi-fi audio signals and digital signals by the three transmitter/receiver pairs 150/170, 178/179, and 200/210 described above.

Communication of signals also takes place in the opposite direction. Specifically, infrared control signals from source 277 are detected by process 274, converted to electrical signals at an RF frequency, and transmitted through filter 271 and coupling network 267. These signals then transmit through filter 265 and across network 264. At video/hi-fi transmitter 250, the control signals are received through filter 261 and transmit through coupling network 260 and filter 259, being received by signal process 256. That component converts the control signals to infrared form, and broadcasts them to receiver 253.

Although some of the functions of the electronic components in Fig. 7 are described above and in U.S. Patent No. 5,010,399, the description is repeated below for easy reference.

Coupling network 260 provides junctions for the signals converging at transmitter 250. Coupling network

267 provides an analogous function in receiver 280. In addition to supplying a simple junction of four paths, each network 260 and 267 also matches the impedances of the various paths, and balances signal energy across the two leads of the transmission line (i.e., the red-green wire pair of network 264 over which the signals are transmitted). Both the impedance matching and signal balancing reduce radiation, while impedance matching makes transfer of energy across the junction that each coupling network 260, 267 introduces more efficient.

One embodiment of coupling network 260 is shown in Fig. 8. The principle element of this network is a transformer wound on a torroid core 260'. There are four isolated windings corresponding to the ports leading to filters 257, 258, 259, and 261. The winding arrangement method shown for the phone line port (in which two lines of the port are connected to a center tap of the winding and interconnected ends of the coil) serves to maximize the balance of signals transmitting on the path leading from that port.

There are different number of windings on the torroid core for the four different ports. (The number of windings shown are only for purposes of illustration.) The turns ratios determine, approximately, the impedance matching between the telephone port and the other three ports. Different ratios will be needed if the telephone line port has a different impedance at the frequencies used for transmission of the various signals.

Coupling networks 260 and 267 can also be designed, using known devices (such as RF splitters and filters) to provide the function of directional multiplexing. This can be used to separate or isolate the three different signals that converge at their ports. These functions and the reason they are required are described in greater detail below.

High-pass filter 261 (Fig. 7), which can be provided

by a single capacitors connected in series along either of the wires of the conductive path, connects between network coupler 260 and the telephone network 264 to block transmission of signals below the highest frequency ordinarily used by telephone equipment. High-pass filter 265 performs an analogous function in receiver 280. This renders connection and operation of the transceivers completely transparent to any low-frequency communications. In U.S. Patent No. 5,010,399, these high-pass filters are included as part of the coupling networks.

Ports 262a, 268a are supplied by transmitter 250 and receiver 280 for connection of telephone equipment. These ports are connected to telephone network 264 through low-pass filters 262 and 268, respectively. Filters 262, 268 prevent telephone equipment 263, 266 from "loading down" any of the signals used by this communication system (i.e., the video, audio, and control signals exchanged by transmitter 250 and receiver 280 over network 264).

As discussed above, signal processing is nearly always required to transform the signals before they are fed to telephone network 264. According to the transmission techniques described above and in U.S. Patent No. 5,010,399, signal processor 254 may modulate, frequency shift, or amplify the video signal it receives as input. Signal processor 255 may perform the same or different processes on the hi-fi audio received from source 252. Finally, processor 274 (in receiver 280) transduces infrared signals from IR transmitter 277 to electrical signals and modulates and amplifies the electrical signal for transmission over network 264.

Similarly, processing of the recovered signals is sometimes needed to convert them to a form expected by the target receiver 275, 276, 253. According to the communication techniques described above and in U.S. Patent No. 5,010,399, processor 272 may demodulate or frequency convert the video signals for television receiver 275 and

it may also perform automatic gain control. Signal processor 273 demodulates and performs AGC on the audio signals destined for speaker/amplifier 276. Finally, signal processor 256 (in transmitter 250) also demodulates its input, amplify the resulting signal, and convert it to infrared light for use by infrared receiver 253.

The details of the processes that apply to hi-fi signals and some of those that apply to video signals are described earlier in this document. Other details that apply to video signals and details that apply to control signals from infrared transmitters are described in U.S. Patent No. 5,010,399. It will be apparent to those skilled in the art that providing for communication of digital signals within this system can use the signal processing described in connection with transmit/receive pair 178/179 described earlier in this disclosure.

Filters 257, 258, 259, 269, 270, and 271 provide frequency separation and isolation between the video, audio, and control signals, and between this group of signals and the telephone signals present on active telephone network 264. If, for example, a video signal transmits within the band spanning from 24 MHz to 30 Mhz, the passband of filters 257 and 269 will cover those bands. If audio signals transmit within the band from 45 to 50 Mhz, the passband of filters 258 and 270 covers those frequencies. Finally, if the control signals transmit within a narrow band centered at 10.7 MHz, filters 271 and 259 will be passband filters centered at that frequency. The specifics of these functions are described in the following paragraphs. An explanation of how directional multiplexing in the coupling networks can also provide some of these functions is presented after that.

Each of the filters 257, 258, 259, 269, 270, and 271 are applied across one of the two-wire paths leading from a coupling network 260, 267 towards one of the sources or receivers. These filters will attenuate signals at

frequencies outside the band of the signal intended to cross that path. Although this filtering is not always necessary, it can never be functionally harmful, and it can be important for several reasons.

First of all, filtering can prevent the component opposite the coupling network from loading down a foreign signal, draining energy away from its intended receiver. One example is filter 257, which, by blocking energy at the frequency of the control signals (from IR transmitter 277), substantially prevents loading processor 254 from loading down the energy of those signals. If not for filter 257, processor 254 could attenuate or reduce the SNR of the control signal incident at processor 256 below adequate levels. Filter 258 functions in a similar way to substantially prevent loading of the control signals by processor 255, and filter 271 substantially prevents loading of the video signal or the hi-fi signal by processor 274.

Filtering can also prevent receivers or processing components from reacting to signals other than the signals of interest. If filter 259, for example, has a narrow passband centered at the frequency of the control signals, it will prevent the video signal and the hi-fi signal from reaching signal processor 256. Because the video and hi-fi signals are being transmitted onto telephone network by transmitter 250, they are at a much higher energy level than the recovered control signals for infrared receiver 253, and would ordinarily disrupt processor 256. Similarly, filter 270 may be necessary to prevent the control signals from disrupting the operation of processor 273.

As described in U.S. Patent No. 5,010,399, processor 272 can include a video channel conversion possibly followed by AGC, or automatic gain control. Neither of these processes, however, are likely to be affected by the control signal energy. Furthermore, control signal energy

reaching the TV is not likely to be a problem because televisions suppress energy at frequencies other than the ones to which they are tuned. The attenuation of the control signals from IR transmitter 277 provided by filter 269, therefore, may not be important.

Finally, filtering can prevent energy generated by a source 251, 252, 277 at out-of-band frequencies from reaching the rest of the system. Effectively, this "cleans up" the respective source signals. An example is filter 271 which, by having a passband cutoff slightly above the fundamental frequency of the control signals produced by processor 274 can block harmonics of such signals. This will be important if the harmonics include energy at the same frequencies over which video signals (destined for receiver 275) or the audio signals (for use by speaker/amplifier 276) are transmitted. For example, if processor 274 generates a control signal centered at 10.7 Mhz, it is likely to have significant harmonic energy at 21.4 Mhz. If filter 271 is a low pass filter with a cutoff of 15 Mhz, it will pass the fundamental of the control signal, but not the 21.4 Mhz harmonic. This will prevent that energy from interfering with reception, by process 272, of video signals covering that frequency.

Processor 254 includes a video modulator and processor 255 includes a hi-fi modulator. Thus, they are likely to include filters that suppress out-of-band frequencies internally.

As discussed above, coupling networks 260 and 267 can provide directional multiplexing to achieve some of the isolation described above. Specifically, coupling network 260 can be designed to isolate the three paths leading to the video, audio, and control processors. This will substantially prevent the video and audio signals from being applied to, and possibly interfering with the operation of, control signal processor 256.

Fig. 9 shows a design that will accomplish this

isolation, as embodied in network coupler 260. The signals from video processor 254 and the signals from hi-fi processor 255 are applied to the inputs of splitter 281. Splitter 281 couples both input signals to splitter 282 through line 281a, but does not permit either input signal to cross over between the video and audio paths. That is, splitter 281 provides a high degree of isolation between paths 254a and 255a. Another port of splitter 282 is coupled through line 256a to control signal processor 256. The third port of splitter 282 is connected through line 282a, through balancing and impedance matching circuitry 282b to high-pass filter 261. Splitter 282 allows the combined video and audio signals to flow through to filter 261 and onto telephone network 264, but prevents crossover of those signals to line 256a that carries the control signals. Control signals transmitting from filter 261 pass through splitter 282, with half of the energy transmitting towards control signal processor 256, and the other half transmitting towards the other processors 254, 255. Thus, the control signals suffer approximately a 3db loss due to the split. Processors 254 and 255, however, are substantially prevented from loading down the control signal energy.

Coupling network 267 can provide similar directional multiplexing. Specifically, coupling network 267 can isolate the three paths leading between the telephone network 264 and the video, audio, and control processors 272-274 in receiver 280. This can prevent the control signal from being applied to signal processors 272, 273, and can prevent processor 274 from loading down the video or audio signals.

The embodiment of coupling networks that perform the balancing and impedance matching described above is described in detail in U.S. Patent No. 5,010,399 for the case of RF video signals and RF modulated control signals. Techniques to extend those networks to include other RF

signals is a procedure that will be apparent to those skilled in this technical field. Techniques to include the directional multiplexing in these networks are also well known.

Although the transmitter 250 and receiver 280 transmit video and audio in one direction and control signals in the reverse direction, these devices demonstrate the principles of transmission and reception of multiple RF signals by a single device that connects to active telephone wiring. Using these techniques, those skilled in the art can design devices that transmit and receive any number of video, audio, and other sources (such as digital sources). The limits on the number of signals will be dictated by governmental limits on electrical radiation and the increasing attenuation of the wiring as higher frequencies are used.

Part II - Cable Television Distribution and Communication System Utilizing Internal Telephone Lines

General Overview (Fig. 11)

Referring to Fig. 11, an interface 300 between public telephone network 301 (i.e., telephone lines located outside of a residence) and a network 302 of telephone wires disposed internally within the residence includes a low-frequency signal processor 311 and an RF/Video processor 312 that communicate signals across residential active telephone wiring network 302 under the control of master controller 316. Processors 311, 312 and master controller 316 are included (together with filters 313 and 314) within the same housing, indicated by interface box 300, and connect to network 302 at the same point. This allows easy communication between processors 311, 312, and master controller 316.

Video transmitters 304a-304c and video receivers 303a-303d are connected to residential telephone network 302 via individual telephone jacks (also called nodes) and transmit and receive video signals and infrared control signals as described in U.S. Patent No. 5,010,399 and Part

I of this disclosure. In the example shown, video receivers 303a-303c are connected between network 302 and televisions 305a-305c, respectively, and video receiver 303d is connected between network 302 and VCR 307. Video transmitters 304a-304c are connected between telephone wiring 302 and camera 306, video game 308, and VCR 307, respectively. Low-pass filters 309a-309c are disposed between respective telephone devices 310a-310c and the jacks leading to network 302 to block high frequency signals, such as switchhook transients, that may be generated by the telephone devices. Filters 309a-309c also prevent telephones 310a-310c from loading down the energy of video signals transmitting across the wiring. This possibility exists because video signals and telephone signals are all transmitted over the same pair of wires, such as the red-green pair, of network 302. This type of communication system is described in detail in U.S. Patent No. 5,010,399 and Part I of this disclosure.

High pass filter 313 and low pass filter 314 separate the signals transmitting between network 302 and interface 300. High pass filter 313 ensures that only video signals (i.e., signals transmitted from video transmitters 304a-304c to RF/video processor 312 or those transmitted by RF/video processor 312 to video receivers 303a-303d) pass between network 302 and RF/video processor 312. Low-pass filter 314 allows only low-frequency signals (e.g., the telephone signals) to pass between low frequency processor 311 and network 302. RF/video processor 312 also receives incoming signals from coaxial cable 315a that connects to port 315 on interface 300 (cable 315a receives cable TV signals from outside the residence), and exports video signals through port 321 on interface 300 for purposes described below.

RF/video processor 312 transmits video and control signals onto network 302 and simultaneously recovers both types of signals from network 302. The video signals it

receives are those transmitted by video transmitters 304a-304c. The video signals RF/video processor 312 transmits may be those provided by cable 315a that connects through port 315, or they may be those it receives from network 302 and retransmits onto network 302 at different frequencies, all as described below.

The ability of processor 312 to retransmit received video signals allows video transmitters 304a-304c and video receivers 303a-303d to communicate video information by relaying such signals through RF/video processor 312. That method has some important advantages, described later on, over direct transmission of signals from the video transmitters (e.g., transmitter 304a) to the video receivers (such as receiver 303b). Communication by direct transmission (e.g., directly between transmitter 304a and receiver 303b) is, of course, also possible.

In addition to the electronic functions of the various components, coordination of: 1) the channels used for transmission by RF/video processor 312; 2) the channels chosen by transmitters 304a-304c; and 3) the specific path, either direct or retransmitted, have a large affect on the viewing options and conveniences available at televisions 305a-305c. Several systems of coordinating these functions are described herein, each of which has particular advantages.

Low frequency processor 311 connects to network 302 via low pass filter 314. This prevents processor 311 from loading down high frequency (e.g., video) signals on network 302. As discussed below, processor 311 detects touch tone (DTMF) signals from any telephone 310a-310c on internal network 302, detects DTMF signals transmitting from network 302, senses incoming telephone calls (i.e., the ringing signal) from public network 301, and passes telephone signals between telephone networks 301, 302.

Master controller 316 achieves two-way communication with processor 311 and RF/video processor 312 via

respective sets of communication links 311′ and 312′. (Links 311′, 312′ each include multiple individual wires.) In addition to the data applied over these links (described below), links 311′, 312′ allow controller 316 to direct the operation of processors 311, 312. Standard digital communication links are adequate for links 311′ and 312′.

Master controller 316 also communicates, via link 311′, with a modem 317 that is disposed within low frequency processor 311 and connected to public network 301. Effectively, this allows master controller 316 to transmit and receive digital signals over public telephone network 301.

Master controller 316 operates under the control of the user. The user can exercise this control by communicating with controller 316 through keypad/display 320 (which is, e.g., mounted on the exterior of a housing that contains interface 300), which connects to controller 316 via link 316′. Communication port 319 also provides communication with controller 316. It conducts communication over link 319′ using standard techniques such as the IEEE RS-232 standard. This allows many different digital devices to connect to controller 316. As described in detail below (see Fig. 18), master controller 316 includes a processor and a digital memory to allow the user to program controller 316 with a set of commands using standard digital programming techniques.

The user can also communicate with and exercise control of master controller 316 by applying DTMF signals onto network 302 from any telephone 310a-310c. The DTMF signals pass over network 302 in the ordinary manner, and are received by processor 311. (As is known, DTMF signals have frequencies within the telephone voice band, such as 4000 Hz or below, and thus are passed by low pass filter 314.) Processor 311 detects the DTMF signals and converts them to digital signals using any suitable technique. The digitized signals are relayed to processor 316 via

communication link 311′. The operation of controller 316 in response is discussed in more detail below.

The user can also communicate with and exercise control of master controller 316 via control signals from infrared (IR) transmitters (such as an ordinary, hand-held IR remote control unit 307′) that are transmitted to RF/video processor 312 via telephone wiring 302. As discussed in U.S. Patent No. 5,010,399 and Part I of this disclosure, the IR signals are detected by a video receiver (such as video receiver 303a), converted to electrical signals at frequencies that differ from both the video signals and the telephone signals on network 302, and transmitted onto the network 302. The control signals are detected by RF/video processor 312 and converted to digital signals (also using techniques that are described in U.S. Patent No. 5,010,399 and Part I of this disclosure), and are transferred to master controller 316. The details of how master controller 316 responds to the control signals are described below.

Master controller 316 communicates with the user by providing textual and other graphics overlaid on the video signals distributed through the residence and viewed on televisions 305a-305c. This is done with graphics processors 329a-329e (Fig. 12) in a manner discussed in more detail below. Master controller 316 also directs RF/video processor 312 to transmit electrical versions of the infrared control signals that control the infrared responsive devices in the residence. Examples of this capability are also described at greater length below.

<u>RF/Video Processor 312 (Fig. 12)</u>

RF/video processor 312 improves the video transmission system by recovering video signals transmitted on network 302 by video transmitters 304a-304c, and retransmitting the recovered video signals to receivers 303a-303d. RF/video processor 312 also selectively transmits externally provided signals available at port 315

over network 302 together with (or in place of) the retransmitted video signals. As discussed, the signals applied to port 315 typically are incoming cable TV signals provided by a local cable TV company, but of course any video signals fed to port 315 can be used.

The signals transmitted by processor 312 onto network 302 are recovered by video receivers 303a-303d, converted to ordinary television signals, and passed to connected televisions 305a-305c, and VCR 307, respectively. To implement these functions, RF/video processor 312 works together with controller 316 to perform the following:

1) Simultaneous reception, via telephone wiring 302, of video signals from video transmitters 304a-304c and of control signals from IR control devices (e.g., remote control 307') via video receivers 303a-303d.

2) Conversion of externally provided video signals (e.g., cable TV signals applied to port 315) to the waveform, frequency, and energy level at which they will be efficiently transmitted across telephone wiring 302.

3) Conversion of video signals received from transmitters 304a-304c on network 302 in one frequency band to a different frequency band and a higher energy level (and possibly to a different form of modulation) for retransmission to video receivers 303a-303d on network 302.

4) Transmission of stored commands that correspond to electrical versions of the infrared signals transmitted by control devices (such as remote control 307') used to control infrared responsive video sources connected to network 302. These signals are transmitted in electrical form across network 302 to video transmitters 304a-304c which, in turn, recreate the infrared patterns that control the devices and broadcast the IR signals for receipt and response by VCR 307 and other infrared responsive devices.

Step 1 is thoroughly discussed in U.S. Patent No. 5,010,399. U.S. Patent No. 5,010,399 and Part I of this disclosure also describe the relationship between signal energy, frequency band, and waveform and the ability to

transmit across the telephone wiring. Described in this section are several techniques for assigning frequency bands (i.e. channels) to the various signals that are sent over network 302, and details of several devices that implement these techniques.

As shown in Fig. 11, RF/video processor 312 connects to network 302 via the red/green wire pair of the telephone line, which is the preferred pair for transmission of RF frequencies. (The black/yellow pair, of course, could be used instead of the red-green pair). Wire pair 313a, which connects to the red-green wire, leads through high-pass filter 313, which blocks only low-frequency (i.e., telephone) signals, allowing RF/video processor 312 to receive all RF signals fed onto network 302 at other locations.

RF/video processor 312 can convert a video signal that it recovers from a video transmitter 304a-304c to a different frequency band (and possibly also to a different form of modulation, such as AM or FM) and retransmit the converted video signal over network 302. The retransmitted signals are then recovered from network 302 by video receivers 303a-303d. There are several possible advantages to this. First of all, video signals that are transmitted in frequency modulated (FM) format by video transmitters 304a-304c can be converted to AM (amplitude modulation) by RF/video processor 312 before retransmission. This confines all conversion circuitry to a single device, which should be somewhat less expensive than providing conversion circuitry in each of receivers 303a-303d. (Conversion is required because typical televisions can only tune AM.) Secondly, if all signals transmitted by video transmitters 304a-304c are retransmitted by RF/video processor 312, all signals reaching any video receiver 303a-303d originate from the same point. This can also simplify the design of the receivers 303a-303d, as will be shown later. Finally, retransmission places all video signals under the control

of a common processor -- RF/video processor 312. This allows a single device to perform additional processing, such as inserting graphical overlays, on any signal, as described later on.

Referring also to Fig. 12, details of RF/video processor 312 are now described. Fig. 12 also shows residential telephone network 302 (as a single block) and master controller 316. All other major components shown are part of RF/video processor 312.

Coupler 325 provides the interface to network 302 via high pass filter 313. Coupler 325 includes five ports, and allows free passage of signals in all directions. It also matches the impedance of each port to that of the telephone line, and balances the signals transmitting onto network 302 across the two conductors of that line. (A similar coupling network that includes four ports is shown in Fig. 8 of Part I of this disclosure.) Extension of that network to provide the function of this network is believed to be a straightforward procedure to those of ordinary skill in this technical field.)

Because coupler 325 allows free passage of signals in all directions, filtering at its ports determines which signals are transmitted along the various paths leading from coupler 325. This is possible because the RF signals generated by transmitters 304a-304c ("video in") and the RF signals transmitted by subprocessor 337 ("video out") are all in different frequency bands; in addition, the "video in" signals and the "video out" signals are at frequencies that are outside of the frequency band that contains the control signals broadcast by receivers 303a-303d ("control in") and the control signals generated by processor 330 ("control out"). Thus, bandpass filter 335 passes the "video in" signals to demodulators 326a, 326b while rejecting the other three signals just described. Similarly, bandpass filters 334, 336 route the "control in" and "control out" signals to and from control signal

processor 330. Finally, bandpass filter 333 passes only the "video out" signals intended for transmission onto network 302 from subprocessor 337.

The signal path leading from filter 335 to demodulators 326a, 326b splits so that demodulators 326a, 326b each receives all of the RF signals recovered from network 302. Cable TV signals provided at port 315 are split three ways, and are applied to demodulators 326c-326e. Amplifiers (not shown) may be provided before splitting to compensate for the splitting loss and ensure that all demodulators 326a-326e receive input signals at sufficiently high SNRs.

Demodulators 326a-326e comprise the first stage of selection and conversion subprocessor 337, the primary functions which are to: 1) select several video signals from among those received from network 302 and those provided by cable 315a; 2) descramble, if necessary (with demodulators 326a-326e), each of the selected signals; and 3) convert the selected signals to the waveform, frequency band, and energy level at which they will be efficiently transmitted across network 302 to video receivers 303a-303d.

In other words, two communication lines apply input signals to subprocessor 337: the red/green twisted pair 313a of the telephone line and the incoming coaxial cable 315a that connects at port 315. Each of these lines provides multiple video signals at different channels. Subprocessor 337 selects some of these signals, combines them onto a single path at different channels, and provides the combined signals as output (called "video out" in Fig. 12).

Subprocessor 337 includes two additional functions. One of these functions is to alter video signals by overlaying graphics on selected signals passing through its circuitry. Graphics processors 329a-329e are used for this purpose. The other secondary function is to provide video